Your Guide to Troubleshooting VoIP

VoIP's extreme sensitivity to delay and packet loss compared to other network applications such as web and e-mail services, presents a real challenge. A basic understanding of VoIP traffic and of the quality metrics provided by VoIP monitoring tools will help keep your network running smoothly.

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NETWORK® INSTRUMENTS

Your Guide to VoIP

This white paper guides you through the essentials of VoIP troubleshooting, including common problems and the metrics you should employ to fix and prevent them. You'll learn how these metrics relate to one another and to overall VoIP health. You'll also learn the best practices for keeping your VoIP network flowing smoothly, your time-to-resolution low, and end users happy.

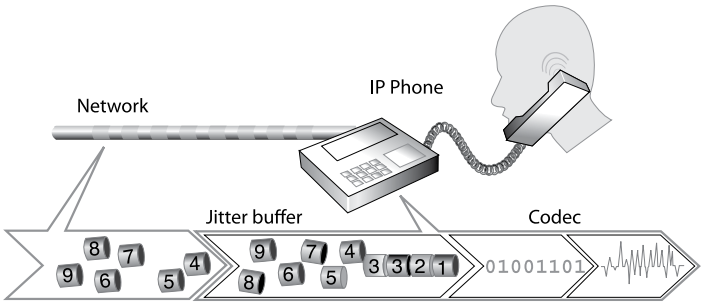
Network Jitter and Delay

Real-time voice communications are sensitive to delay and variation in packet arrival times. Codecs require a steady, dependable stream of packets to provide reasonable playback quality. Packets arriving too early, too late, or out of sequence result in jerky, jumbled playback. This phenomenon is called jitter.

Increasing jitter buffer size can help, but only to a point.

Because no network can guarantee a perfectly steady stream of packets under real-world conditions, VoIP phones use jitter buffers to smooth out the kinks. A jitter buffer is simply a First-In, First Out (FIFO) memory cache that collects the packets as they arrive, forwarding them to the codec evenly spaced and in proper sequence for accurate playback.

Jitter buffering and packet loss concealment



VoIP packets can arrive at the receiving phone out of sequence, late, early, or not at all. IP phones use a *jitter buffer* to reconstruct the packet stream at the receiving end, duplicating missing packets or filling in with white comfort noise when necessary.

While a jitter buffer can successfully mask mild delay and jitter problems, severe jitter can overwhelm the jitter buffer, which results in packet loss (see below). Increasing the size of the jitter buffer can help, but only to a point: A jitter buffer that increases overall round-trip delay to 300 ms will make normal conversation difficult.

Packet Loss

As mentioned above, packet loss can be the result of the jitter buffer being overwhelmed. Other reasons include landline media failure and poor wireless signal quality. The latter can be a big problem with VoFi (Voice over WiFi) service. Regardless of the source, VoIP phones and gateways attempt to conceal this type of signal degradation by duplicating packets to fill in the missing data. As with jitter, these techniques can maintain voice quality only to a point.

Packet loss on data networks has long been characterized as a "bursty" phenomenon, which is another way of saying "it never rains, it pours." Networks tend to either sporadically drop single packets (these periods are called "gaps" in packet loss), or large numbers of contiguous packets in a "burst." Packet loss concealment techniques typically have no problem handling packet loss during gap periods; it is the sustained bursts you must watch out for.

Call Management Problems

If the VoIP call manager (sometimes called the VoIP server) is overwhelmed with requests, or its connection to the network is impaired, call setup delays can reach the point where users abandon calls before they are able to connect to the other party. If IP phones are misconfigured, or their IP connection to the server is impaired, calls remain open in the call queue long after the parties have disconnected.

A major Midwest health provider recently undertook a major VoIP deployment.

The hospital used Observer® Expert to conduct network assessments to benchmark network performance before implementing VoIP.

"We were in the middle of a significant VoIP deployment and had plans for implementing other critical network applications.

We really needed some type of network analyzer to monitor and maintain network performance."

Jack King,
Director of I.T.

Managing VoIP Quality

You can manage only what you can measure. Managing a VoIP deployment therefore requires some hard numbers beyond subjective user assessments of quality (although these are obviously important as well). Beyond monitoring the network parameters discussed in this paper, having an overall quality score such as a Mean Opinion Score (MOS) or R-factor score can also be a useful VoIP network health index.

VoIP monitoring tools calculate the MOS and R-factor scores using a formula known as the E-model. Using the statistics it has collected from the network, the analyzer calculates how much the various impairment factors (such as codec compression, jitter, delay, and packet loss) would affect the typical user's perception of call quality.

VoIP-Specific vs. All-Purpose Monitoring Tools

There are a number of different options on the market for managing VoIP quality, mainly falling into three categories:

- Dedicated VoIP tools originally developed for the telecom industry. These tools are great for testing IP phone and gateway designs, but not as good at solving deployment problems on a live network.
- Network protocol analyzers that have added "VoIP Support" by licensing technology developed for the telecom industry and integrating it into their product line.
- Network monitoring tools that approach VoIP quality management from an IT administrator's point of view rather than from that of telecom engineer.

To the IT administrator, managing VoIP quality is just another network task. This makes the third approach (the "all-purpose" network monitoring tool) often the most practical choice. But note that "VoIP support" means more than just decoding the packets of various VoIP protocols; it also means being able to track and display network delay, jitter, and packet loss, and to distill this information into overall quality scores, both per-call and in aggregate.

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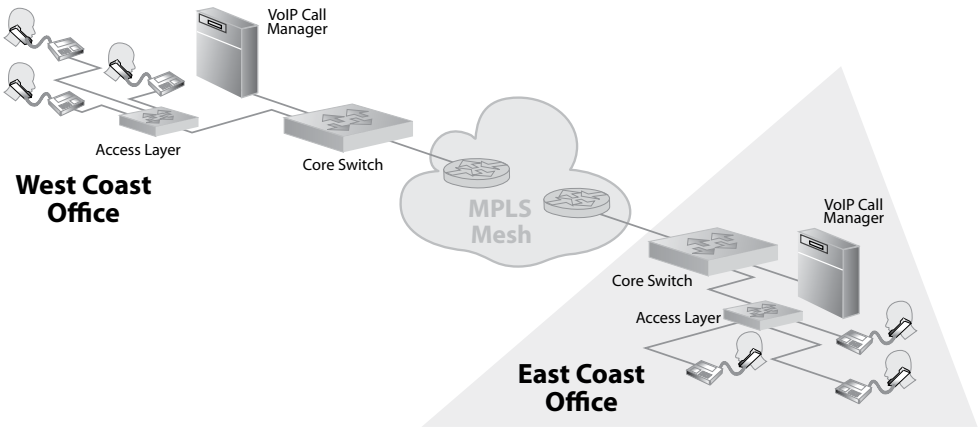
And to be really useful to the enterprise, the tool should also track, store, and analyze long-term trends. This is so that you can understand what is "normal" VoIP performance, and maintain a database of Call Detail Records (CDRs) from which you can generate reports for management or service providers. The VoIP monitoring tool should also be capable of automatically notifying you when selected statistics indicate a developing problem. On all of these counts, Network Instruments® Observer® meets the requirements.

VoIP Points of Visibility

In switched environments, where to deploy an analyzer or probe for maximum visibility isn't necessarily obvious. Complicating matters for VoIP is the fact that each call includes both client-server communications (between IP phones and the call manager during setup and tear-down), and peer-to-peer (the streams of voice data passed between the parties). For example, consider the following VoIP network deployment:



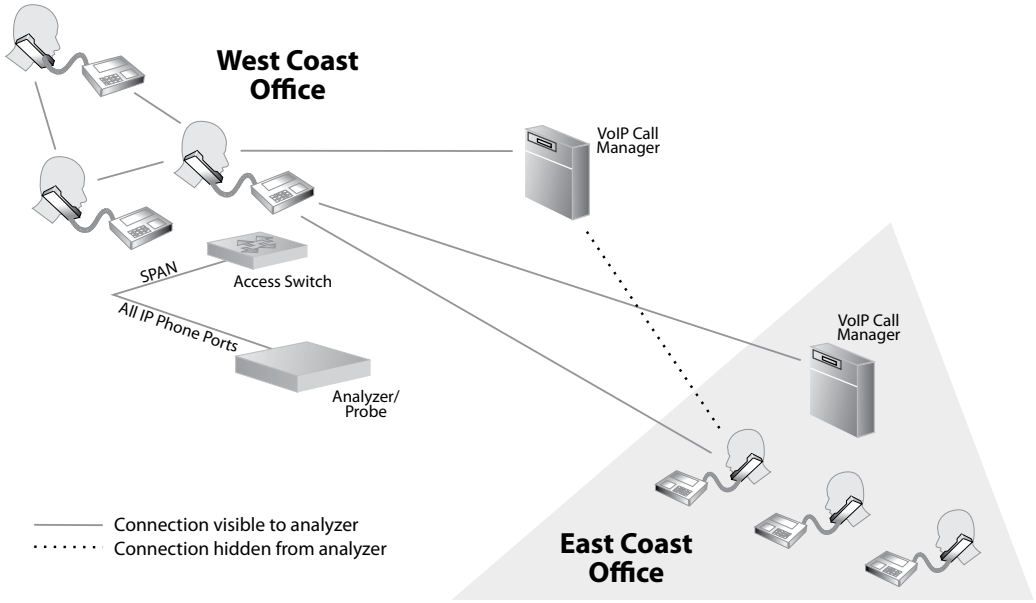
How Call Quality Relates to User Satisfaction

MOS and R-factor are used to gauge user satisfaction with call quality. MOS levels under 3.5 and R-factor below 80 mean trouble.

Where to place probes on such a network depends on what you want or need to see. If you need access to all local conversations on either coast, including both call setup and actual voice data, use a SPAN session on the access layer switch to mirror VoIP traffic to the analyzer. Assigning all VoIP traffic to a dedicated VLAN makes this fairly straightforward.

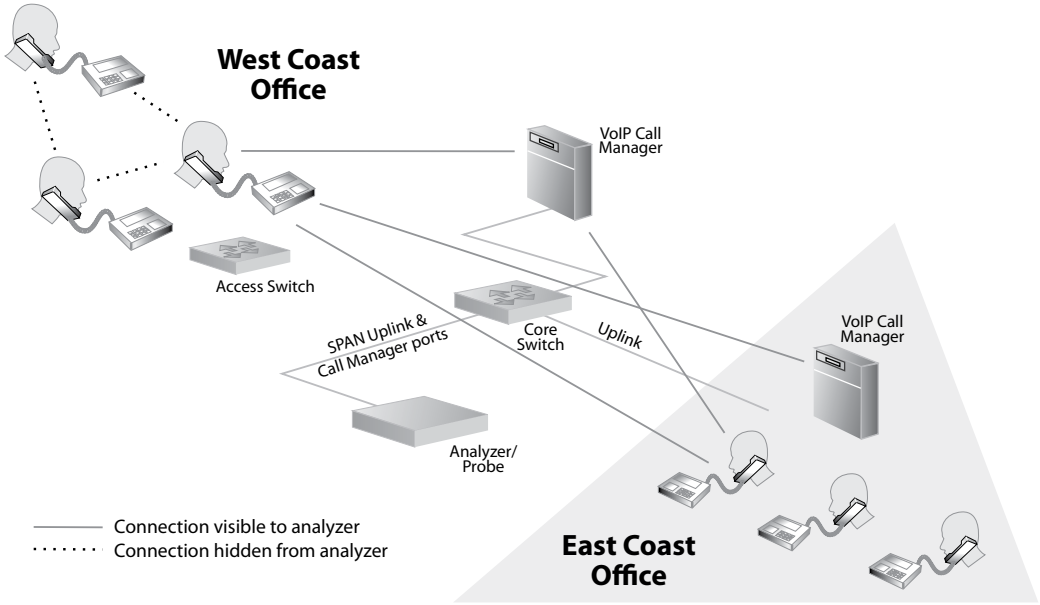
Capturing Local IP Phone Traffic Shows:

- Any phone's communications with its local call manager
- Both sides of the full-duplex connection between local phones talking to each other
- Both sides of the full-duplex connection between phones located on opposite coasts



What you will not be able to see from this probe is any communications between the East Coast and the call manager located on the West Coast.

If you are more interested in a coherent view of calls between the West Coast and East Coast, including all call manager communications, use a SPAN session to mirror both the uplink traffic between the core and MPLS mesh, and all traffic flowing to and from the call manager. This will give you a coherent view of inter-office calls, along with all call manager communications, both local and remote.



"Observer wins hands down."

Christian Wilson,
Network Administrator
Select Comfort

With a probe deployed in this manner, you will not be able to see the peer-to-peer voice traffic between local phones. For complete coverage, connect probes to both the core and access layers at each site. Another alternative is to deploy probes at the core 24/7/365, monitoring the access layer with a portable analyzer or software probe only to troubleshoot local call problems as needed.

VoIP Network Analysis

How can VoIP analysis help manage quality? By closely monitoring the network conditions that affect VoIP, you can begin to address developing infrastructure problems before they result in user complaints or downtime.

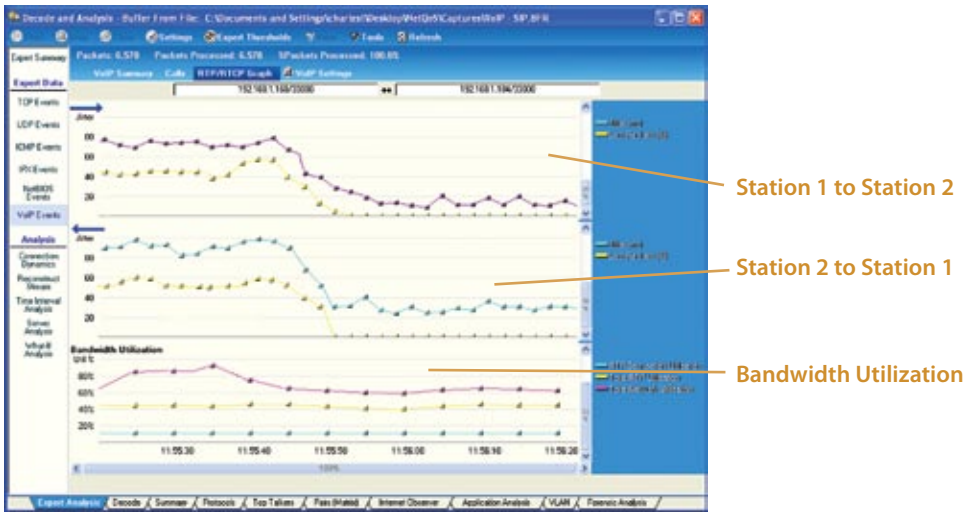
Tracking Network Performance

Consider the East Coast/West Coast example described in the previous section. Ken, the administrator responsible for ensuring VoIP quality, has set up a Network Instruments 10/100/1000 Probe Appliance on the core switch to monitor all call manager activity and any VoIP traffic traversing the link. He has configured Observer to send him an e-mail whenever any of the following conditions arise:

- MOS falls to 3.5 or less
- Jitter levels crossing the MPLS mesh exceed 20 ms
- Delay levels crossing the MPLS mesh exceed 80 ms

Any of these conditions are indications that VoIP quality is threatened. Given the topology involved, the most likely source of problems is the MPLS mesh routers, which are under the service provider's control. By digging deeper into the statistics the analyzer provides, you can determine why the MOS is falling, and what is causing jitter, delay, or packet loss.

If jitter is the problem, a good place to start is by comparing jitter levels against bandwidth utilization to see if there is any correlation. The analysis shown below (taken from Network Instruments' Observer Expert) shows just such a correlation.



Comparing Jitter to Bandwidth Utilization

As this scenario shows, such a link between spikes in bandwidth and jitter could mean it is time to invest in more bandwidth, or time to put more controls on employee Internet usage for applications such as streaming media and peer-to-peer file sharing unrelated to business.

If there isn't an obvious correlation between jitter and bandwidth utilization, the depth of data provided by an all-purpose network analysis and monitoring tool can help you dig deeper for the correct diagnosis. For example, if VoIP traffic across an MPLS mesh is subject to excessive jitter, it could be the result of "route flapping" on the service provider's routers. An analyzer can confirm and document this. Armed with the hard data provided by analysis, you could then contact the service provider so they can address the problem. If delay across the mesh exceeds the contractual obligations of the Service Level Agreement (SLA), the provider may owe your organization some refunds for service failure, in addition to being responsible for fixing the problem.

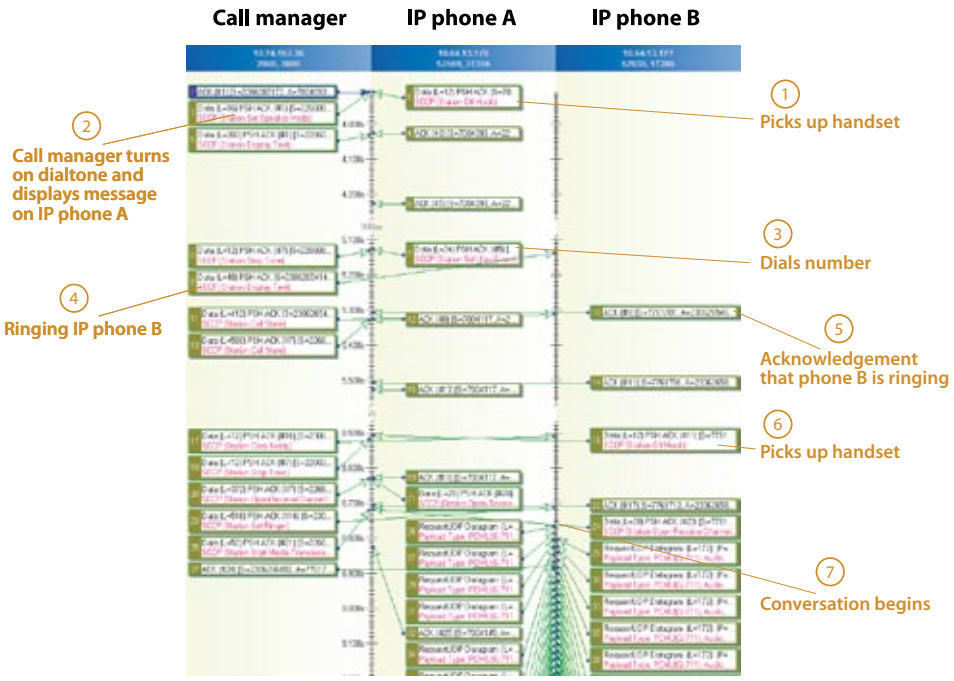
"With Observer it's like looking in 60 directions at once."

Coleman Jennings,
Senior Network Engineer
AAA East Central

Troubleshooting Connection Problems

When a user can't get a dial tone, or if there are excessive delays in ringing the other party's phone, examining a graphical display of how the call is progressing between the parties and the call manager can indicate what is going wrong.

Network Instruments' VoIP Expert displays just such a diagram: simply right-click on any call or connection stream. Because differing protocols dictate differing phone/call manager interactions, some knowledge of the protocol is necessary for detailed troubleshooting. But even if you lack a detailed knowledge of the protocol, the Connection Dynamics display highlights which party isn't responding, or which party is responding slowly.



An example of a Connection Dynamics display showing a VoIP call using the SCCP protocol. It is easy to see how such a diagram is essential to efficiently troubleshoot VoIP connection problems.

Better manage and troubleshoot VoIP across your network using Observer's **VoIP Expert**, available in Observer Expert and Observer Suite. Whether you're interested in the big picture or a specific conversation, Observer offers real-time statistics, Expert VoIP analysis, and reports to help you address all levels of VoIP traffic.

Over 70 VoIP-specific metrics.

Understanding VoIP performance is the key to accurate troubleshooting. Observer provides **visibility into the network, application, and VoIP traffic** to quickly resolve issues. VoIP Expert summaries provide easy understanding of overall VoIP network health. Over 70 VoIP-specific metrics like call quality can be tracked on an aggregate or per-call basis. Take advantage of over 50 VoIP Expert events for immediate problem identification. Once you've identified the issue, drill-down on specific conversations for an in-depth view.

VoIP Expert Analysis Features

Convenient, at-a-glance summaries

- VoIP Traffic Summary
- Call Summary
- Voice Quality Scoring
- Precedence (QoS)

"The Network Instruments tools let me reconstruct an entire VoIP conversation and hear it real-time."

"This tool set allows me to have eyes into a customer's network."

Ivan McDuffie,
NEC Unified Solutions

Detailed views with in-depth Call Detail Records

- Addresses
- Status (Open, closed, fail)
- Number of packets, packet bytes, packet loss
- Start time, initial setup duration, duration
- Current jitter, maximum jitter
- MOS, R-factor
- QoS for each call
- Number of packets that arrive out of order
- Detailed analysis for packet loss and delay
- Gap and burst measurements
- Long-term Call Detail Records trending

Quickly Solve Problems with over 50 VoIP Experts

Observer offers over 50 event-based and threshold-based VoIP Experts to immediately flag problems for faster problem resolution. Examples include:

- Alarms for unacceptable jitter level
- Lost packets
- Alterations in the QoS stream

Manage VoIP Audio Quality

Observer displays individual Mean Opinion Score (MOS) and R-factor for individual calls and as an average. Place alarms on this score to proactively manage overall VoIP quality.

Monitor Quality of Service (QoS)

Observer's VoIP Expert reports QoS levels by call, packet, and protocol. This analysis also shows percentage of VoIP utilization compared to other network traffic, allowing you to plan network upgrades.

Evaluate Jitter

Observer's VoIP Expert displays the statistical variance of packet arrival times, known as jitter, measured in timestamp units or RTP time units.

Measure Bursts and Gaps

Observer's VoIP Expert provides comprehensive metrics for monitoring bursts, or periods of high packet loss, and gaps, or periods of little packet loss. Burst and gap percentages quickly indicate the quality of a call.

Track and Decode VoIP and Video

Observer offers complete VoIP and video decodes, including H.323, Session Initiation Protocol (SIP), MGCP, and SCCP (Cisco "skinny"). Observer also supports Avaya CCMS, Nortel UNIStim, and Mitel® systems.

Reconstruct and Review VoIP Calls

Capture and reconstruct calls for playback, so you can hear the same call quality issues experienced by users.

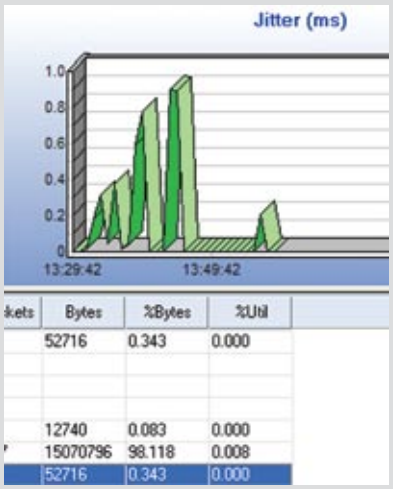
Compare VoIP to Network Performance

Use Observer to identify whether jitter or delay is being caused by other applications on the network. Observer will track VoIP issues along with overall network performance.

Monitor VoWLAN

Observer's VoIP enhancements are automatically available across multiple topologies, thanks to the Network Instruments Distributed Network Analysis (NI-DNA™) architecture. Observer's VoIP Expert will monitor VoIP traffic even over wireless networks.

Observer's VoIP Expert



Plotting jitter in milliseconds.

Top 10 VoIP Best Practices

Our sales engineers have put together this valuable list of the best practices to use when implementing or troubleshooting VoIP. Keeping these steps in mind will help you get the most from your VoIP network.

1. Understand and measure call quality components

There are a variety of metrics you can use to assess VoIP call quality, including jitter, MOS, R-Factor, gap density, burst density, Quality of Service prioritization, and compression techniques. Ensure you are accurately analyzing VoIP communication by learning how to measure these attributes.

2. Implement Quality of Service prioritization

Incorrectly set QoS precedence for VoIP traffic leads to delays in packet delivery and reduced call quality.

3. Conduct site surveys

The more you know about your network, the better prepared you are to properly integrate VoIP. Conduct a site survey to review current WAN bandwidth levels, traffic flows, and existing switches for bottlenecks and choke points. Then, identify or determine specific needs through testing and modeling.

4. Deploy analysis tools strategically for maximum visibility

Placing network analysis consoles and probes on your network requires a clear understanding of VoIP traffic patterns. Are you concerned with monitoring VoIP traffic locally, over WAN links, both? Depending on your objectives, place your analysis tools to ensure optimal visibility of VoIP communications.

5. Implement VLANs to isolate and monitor VoIP issues

Organize your VoIP traffic by VLAN user groups. This practice will greatly simplify problem resolution.

6. Monitor rollouts to ensure a positive user experience

Determine whether users are receiving a positive experience by reviewing cumulative VoIP metrics, codecs, and other network performance variables during VoIP deployment. By evaluating VLAN setups and overall link utilization, you can judge overall network performance and quickly make adjustments during implementation.

7. Compare jitter to overall network bandwidth utilization to understand response time

When jitter becomes a problem, look at the big picture. A correlation between jitter and bandwidth usage means the problem is overall network usage. If there is no direct correlation, excessive jitter might be caused by isolated network factors that require further investigation.

8. Set up your analyzer to proactively monitor VoIP activity

Utilize monitoring and notification tools to speed problem resolution. Determine "normal" or "acceptable" levels of activity for your network and its users. Then set up thresholds within your analyzer to alert you when thresholds are broken or in danger.

9. Automate problem resolution

Expert Analysis functionality eliminates unnecessary trial and error when troubleshooting VoIP issues by automating problem resolution. Utilize Expert Analysis on VoIP communication to quickly pinpoint the source of common VoIP problems.

10. Baseline network traffic

For comprehensive understanding of VoIP traffic, capture and store long-term network data. Only with critical trending data can you accurately perform baselining activities. Baselining validates VoIP performance, helps future capacity planning efforts, and provides long-term understanding of VoIP health.

CI Travel's VoIP Savings

High phone bills were eating up CI Travel's profits. They decided to take advantage of VoIP technology to reduce per-call expenses.

The new VoIP phones, while dramatically reducing per-call costs, came with new problems. To make the investment pay off, Ingram required tools to monitor and troubleshoot the VoIP exchange.

"So far, Observer's VoIP capabilities have helped cut CI Travel's phone bill by 25 to 30 percent."

Paul Ingram,
Director of IT
– CI Travel is a $150 million division of Cruise International.

Summary of VoIP statistics and quality metrics

The following table summarizes the statistics and quality measurements discussed in this paper, both defining what is measured, and describing its relevance.

VoIP metric	What it measures	How to use the analysis
Jitter	Jitter measures the variability of delay in packet arrival times. In spite of the jitter buffers used to counteract jitter, at excessive levels it can interfere with smooth playback and cause packets to be dropped.	By using triggers to notify you when jitter levels are reaching a point that threaten voice quality, you can examine your routers for problems or contact your service provider and help them solve the problem.
Delay	The amount of time it takes a packet to reach its destination. Whenever packets travel a network, some delay is inevitable. For real-time telephone conversations, there is a one-way "delay budget" of approximately 150 ms.	As with jitter, using automatic notifications to actively manage levels of delay can prevent the problem from escalating to the point where users complain.
Packet loss	The percentage of packets that did not reach their destination.	Sporadic packet loss is usually insignificant. However, sustained bursts (see the next item) can affect quality.
Bursts	Periods characterized by high rates of packet loss. The burst percentage is the percentage of time that the call experienced high-rate packet loss; the burst density is the actual percentage rate of packet loss during bursts.	VoIP phones have no trouble masking a lost packet here and there by duplicating the previous packet or filling longer silences with white noise. But users will notice sustained bursts. If VoIP traffic has been assigned proper QoS and has enough bandwidth, the most likely culprit is media failure.
Gaps	Periods characterized by low rates of packet loss. The gap percentage is the percentage of time that the call experienced low-rate packet loss; the gap density is the actual percentage rate of packet loss during the gaps.	Usually not significant, as packet loss concealment technologies are usually successful in masking the effects of low-level packet loss. Contrast with bursts, described above.
Average call setup/teardown	An average of how long it is taking the call manager to open and close calls.	A spike in these statistics can indicate a problem with the call manager or its connectivity to the network.
Codec	The compression/decompression method that was used for the call.	Different codecs are capable of different levels of quality sound reproduction. Higher compression comes at the cost of lower quality, but may be necessary given the bandwidth available to the call. If it seems as if the codecs in use are using more compression than necessary (or not enough) given the amount of bandwidth available, perhaps the VoIP phones can be reconfigured to use a different codec.
Mean Opinion Score (MOS) R-factor	Starting with a theoretical perfect score of 5 (excellent), impairment factors such as codec, delay, jitter, and packet loss are used to calculate how a typical user would rate voice quality. Similar to MOS, this scale ranges from 1-100.	These are useful as quick overall indicators of VoIP health. If the average MOS falls below 3.5, or the average R-factor falls below 80, it's likely that you have more than a few dissatisfied users. If you see these statistics trending downward, it's time to examine more detailed analysis to determine what is going wrong.

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